PATENT
1432.136.101/P33803

~~Description~~

CIRCUIT ARRANGEMENT AND SIGNAL-PROCESSING DEVICE

Background

The invention relates to a circuit arrangement and to a signal-processing apparatus.

For automatic speech recognition purposes, it is known for a Fast Fourier Transform (FFT) to be applied to a digitized input speech signal, for spectral analysis of the input speech signal~~.~~. Features which are used for automatic speech recognition purposes are derived from the power spectrum formed by means of the Fast Fourier Transform (see ~~[1]~~ E.G. Schukat-Talamazzini, Automatische Spracherkennung [Automatic speech recognition], Friedrich Vieweg & Sohn Verlagsgesellschaft, Braunschweig-Wiesbaden, ISBN 3-528-05492-1, Chapters 1 to 3, 1995).

In a Fast Fourier Transform such as this, a time window of predetermined duration is normally used, with one signal element which is represented by the respective time window in each case being subjected to a Fast Fourier Transform~~.~~. This leads to restricted frequency resolution and time resolution.

If, as is normal for speech recognition, only the power spectrum and thus the magnitude spectrum of the respective signal element is used, the time resolution is limited by the duration of the time window that is used~~.~~. This limitation on the time resolution is a factor which leads to a restriction to the performance of already known speech-recognition systems~~.~~. One problem with sound processing systems and the use of a time window such as this of a fixed predetermined size is that, if the power spectrum varies, an error is formed after

reverse-transformation to the time domain, because of the finite nature of the time window.

According to alternative approaches for automatic speech recognition, as described in ~~[2]~~ S. Senneff, A Computational Model for the Perforal Auditory System: Application to Speech Recognition Research, Proceedings of IEEE ICASSP 1986, Tokyo, pages 1983 to 1986, Tokyo, 1986, ~~[3]~~ M. Hunke and T. Holton, Training Machine Classifiers to Match the performance of Human Listeners in a Natural Vowel Classification Task, ICSSLP, pages 574 to 577, 1996 and ~~[4]~~ S. Sandhu and O. Ghitza, A Comparative Study of MEL Cepstra and EIH for Phone Classification under Adverse Conditions, IEEE, pages 409 to 412, 1995, filter banks are used. These filter banks are intended to stimulate the characteristics of the human inner ear.

Furthermore, ~~[5]~~ C., Sumner et al., A Revised Model of the Inner-Hair-Cell and Auditory Nerve Complex, Journal of Acoustic Society of America, Vol. 111, pages 2178 to 2188, May 2002 describes a model of the inner hair cells (IHC) and of the auditory nerves, with the basilar membrane being simulated using different filters and a compression unit, as well as an amplification unit. Furthermore, [5] describes a vesicle pool model.

Furthermore, ~~[6]~~ T. Dau, Modell der effektiven Signalverarbeitung im Gehör, Einblicke [Model of the effective signal processing in the hearing, Insights], No. 29, pages 16 to 18, April 1999 describes a model of the signal-processing in the human hearing, with an acoustic sound signal being used as an input signal according to this model. The input signal is subdivided into a plurality of frequency channels, analogously to the frequency-location transformation in the inner ear, with half-wave rectification as well as low-pass filtering being provided for each channel, an adaptation being provided for amplification of sudden changes in the input signal, and for attenuation of components of the input signal which are essentially constant over time.

~~[7]~~ V. Hohmann, Signalverarbeitung in digitalen Hörgeräten, Einblicke [Signal processing in digital hearing aids, Insights], No. 33, pages 24 to 26, June 2001 ~~D~~describes the phenomenon, which is referred to as "recruitment", as well as the ~~so-called~~ dynamic compression in order to compensate for the recruitment phenomenon~~.~~. The dynamic compression results in a wide sound-level value range, which occurs in the acoustic environment, being "compressed" to a range which can be perceived by human beings~~.~~.

~~[8]~~ H. W. Strube, A Computationally Efficient Basilar-Membrane-Model, Acoustica, Vol. 58, pages 207 to 214, 1985 ~~D~~describes a digital simulation of a one-dimensional long-wave model of the basilar membrane of a human being, using a wave digital filter structure~~.~~.

~~[9]~~ M. A. Ruggero et al., Mechanical Basis of Frequency Tuning and Neural Excitation at the Base of the Cochlea: Comparison of Basilar-Membrane-Vibration and Auditory Nerve-Fiber-Responses in Chinchilla, Proceedings of National Academy of Science USA, Vol. 97, No. 22, pages 11744 to 11750, October 2000 ~~D~~describes a growth function, in which the oscillation of the basilar membrane is described with respect to the sound pressure, measured upstream of the tympanic membrane of an experimental animal.

Furthermore, ~~[10]~~ P. Dallos et al., The Cochlea, ISBN 0387944494, Springer-Verlag, Chapter 6, pages 318 to 385, 1998 describes the biological structure of the inner hair cells~~.~~.

~~[11]~~ DE 691 31 095 T2 ~~D~~describes amplifier circuits, in which the amplifiers are used to improve the comprehensibility for a sound projection system~~.~~. The voltage-controlled amplifiers described in ~~[11]~~ DE 691 31 095 T2 are controlled by a gain control signal which is produced by means of a buffer amplifier and is formed by a combination network from the signal elements, as

produced by means of a filter bank, in different frequency channels~~.~~. The voltage-controlled amplifiers as described in DE 691 31 095 T2~~[11]~~ process four different signals in four different frequency channels~~.~~.

The aim of the circuit arrangement described there is to change the output signal produced by the respective voltage-controlled amplifiers to the same level as the amplitude of the signal in the baseband channel.

For these and other reasons, there is a need for the present invention.

Summary

One embodiment provides a circuit arrangement has a filter bank with a plurality of filter stages and a filter-bank input to which an input signal can be supplied. The circuit arrangement has a plurality of resonator circuits for generation of in each case one output signal element from the input signal, with each resonator circuit in each case being associated with at least one filter stage from the plurality of filter stages, and being coupled to an output of the respective filter stage. Each resonator circuit has a capacitance, an inductance and a resonator output, at which the respective output signal element can be produced. Furthermore, at least one resonator control circuit for open-loop or closed-loop control of the Q-factor of at least one resonator circuit is provided in the circuit arrangement, with the resonator control circuit being designed in such a manner that it provides open-loop or closed-loop control for the Q-factor of the at least one resonator circuit as a function of the time profile of the signal amplitude of the input signal and/or of the output signal element from the at least one resonator circuit.

Brief Description of the Drawings

The accompanying drawings are included to provide a further understanding of the present invention and are incorporated in and constitute a part of this specification. The drawings illustrate the embodiments of the present invention and together with the description serve to explain the principles of the

invention. Other embodiments of the present invention and many of the intended advantages of the present invention will be readily appreciated as they become better understood by reference to the following detailed description. The elements of the drawings are not necessarily to scale relative to each other. Like reference numerals designate corresponding similar parts.Figure 1 illustrates a circuit arrangement according to one preferred exemplary embodiment of the invention.

Figure 2 illustrates a resonator circuit according to one exemplary embodiment of the invention.

Figure 3 illustrates an implementation of the resonator circuit as illustrated in Figure 2, in the form of a wave digital filter.

Figures 4 and 5 illustrate diagrams to illustrate the functionality of the circuit arrangement illustrated in Figure 1.

Figure 6a illustrates a circuit arrangement element according to another exemplary embodiment of the invention.

Figure 6b illustrates an implementation of the resonator circuits illustrated in Figure 6a as wave digital filters.

Figure 7a illustrates a circuit arrangement element according to another exemplary embodiment of the invention.

Figure 7b illustrates an implementation of the resonator circuits illustrated in Figure 7a as wave digital filters.

Figure 8 illustrates a block diagram of a speech-recognition system according to one exemplary embodiment of the invention.

Figure 9 illustrates a circuit diagram of a linear filter bank and of a plurality of resonator circuits associated with individual filter stages in the filter bank according to one exemplary embodiment of the invention.

Figure 10 illustrates a diagram illustrating an excitation pattern for a non-linear basilar membrane model for a 1 kHz tone.

Figure 11 illustrates a circuit diagram of a circuit element which is in each case connected in series with a respective resonator circuit in order to form the circuit arrangement according to one exemplary embodiment of the invention.

Figure 12 illustrates an illustration of modeled nerve action potentials.

Figure 13 illustrates a diagram illustrating different speech-recognition rates for a speech-recognition system according to one exemplary embodiment of the invention, compared with a speech-recognition system according to the prior art.

Detailed Description

In the following Detailed Description, reference is made to the accompanying drawings, which form a part hereof, and in which is shown by way of illustration specific embodiments in which the invention may be practiced. In this regard, directional terminology, such as "top," "bottom," "front," "back," "leading," "trailing," etc., is used with reference to the orientation of the Figure(s) being described. Because components of embodiments of the present invention can be positioned in a number of different orientations, the directional terminology is used for purposes of illustration and is in no way limiting. It is to be understood that other embodiments may be utilized and structural or logical changes may be made without departing from the scope of the present invention. The following detailed description, therefore, is not to be taken in a limiting sense, and the scope of the present invention is defined by the appended claims.

The present invention ~~is based on the problem of specifying~~ provides a circuit arrangement as well as a signal-processing apparatus for provision of features to describe a signal which is supplied to the circuit arrangement or to the signal-processing apparatus, with the features being more robust with respect to any interference noise that occurs~~-~~.

~~The problem is solved by a circuit arrangement and by a signal-processing apparatus having the features as claimed in the independent patent claims.~~

~~Preferred refinements of the invention are specified in the dependent claims.~~

A circuit arrangement has a filter bank with a plurality of filter stages and a filter-bank input to which an input signal can be supplied. Furthermore, the circuit arrangement has a plurality of resonator circuits for generation of in each case one output signal element from the input signal, with each resonator circuit in each case being associated with at least one filter stage from the plurality of filter stages, and being coupled to an output of the respective filter stage. Each resonator circuit has a capacitance, an inductance and a resonator output, at which the respective output signal element can be produced. Furthermore, at least one resonator control circuit for open-loop or closed-loop control of the Q-factor of at least one resonator circuit is provided in the circuit arrangement, with the resonator control circuit being designed in such a manner that it provides open-loop or closed-loop control for the Q-factor of the at least one resonator circuit as a function of the time profile of the signal amplitude of the input signal and/or of the output signal element from the at least one resonator circuit.

A signal-processing apparatus has a circuit arrangement as described above as well as a further-processing unit for further-processing of the signal which is produced by the circuit arrangement.

The combination of a filter bank, preferably of a linear filter bank, with the resonator circuits which as can be seen form non-linear compression stages simulates, according to the invention, the non-linear oscillation response of the inner ear of mammals, very well.

Features in the course of a feature extraction process on an input speech signal are as can be seen produced as the output signal of each resonator circuit, which features are more robust to inference noise and, in particular for the purposes of a speech-recognition system, this leads to an improved word error rate in a feature extraction system (formed by the circuit arrangement) provided according to the invention.

In particular, as the interference noise increases, the word error rate increases more slowly than in the course of the traditional initial processing using a Fast Fourier Transform, applied to power spectra which have been split on a time-window basis.

A further advantage which is achieved by the invention is the large-scale maintenance of the fine time structure of the recorded speech signal, in general of the analog input signal which is supplied to the circuit arrangement, while, in the case of the power spectrum when using a Fast Fourier Transform, all that is achieved is time resolution of the features in the region of the window length of the window that is used.

Thus, according to the invention, the biological structure and, in particular, the essential characteristics of the human hearing system are, as can be seen, simulated in a better manner than in the prior art for the purposes of feature extraction of features in an acoustic signal. This leads to a more robust speech-recognition system.

The invention can thus clearly be seen in the extraction of features for, for example, automatic speech recognition, that is to say for an automatic speech-recognition system. In particular, features are produced from an input signal which is supplied to the circuit arrangement, which features allow more robustness to interference noise than can be achieved according to the prior art.

Alternatively, the invention can be used highly advantageously for a hearing aid, for example for a cochlea implant, particularly in the case of patients with impaired hearing caused by the inner ear.

According to the invention, the Q-factor of the respective resonator circuit is adjusted on the basis of the amplitude of the input signal or output signal element. If the amplitude of one of these signals is very high, then the resonator

control circuit can be used to reduce the Q-factor of the at least one resonator circuit to such an extent that the signal is highly attenuated. In contrast, in the case of a low-amplitude signal, the Q-factor can be increased to such an extent that the signal amplitude at the output of the respective resonator circuit is amplified.

As can be seen, according to the invention, in order to carry out dynamic compression, use is made of the fact that a resonator circuit acts a stable amplifier when close to its resonant frequency (resonance peak).

The Q-factor of the resonator circuit used here is understood to mean the ratio of the amplitude of the output signal at the resonant frequency of the resonator circuit to the corresponding amplitude of the input signal. The Q-factor of a resonator circuit depends on its resistance, so that the Q-factor can be adjusted, for example, by means of open-loop or closed-loop control of the resistance of the resonator circuit.

In a scenario in which the Q-factor of the resonator circuit is adjusted on the basis of the amplitude of the input signal that is input in the resonator circuit, the functionality of the control circuit can be referred to as "open-loop control". If, in contrast, the Q-factor of the resonator circuit is adjusted on the basis of the amplitude of the output signal, then the resonator control circuit carries out a "closed-loop control" functionality, since it uses feedback to adapt the Q-factor.

The circuit arrangement according to the invention allows reliable and effective dynamic compression of an input signal in the time domain, without the disadvantages of a Fourier Transform occurring. In particular, this avoids the problems of a finite time window that occur with a Fourier Transform according to the prior art. Furthermore, according to the invention, a dynamically compressed output signal is generated which, for example, has considerably less interference signal distortion than that with the reverse transformation of the logarithmic Fourier spectrum.

According to the invention, sufficiently powerful and intensity-selective (for example non-linear) attenuation of an input signal is made possible by selectively producing the Q-factor of the resonator circuit.

As can be seen, the circuit arrangement has a filter circuit, with the frequency range for which the resonator circuit passes signals being defined on the basis of the value of the inductance L and of the capacitance C of the resonator circuit. The values L, C can thus be set to create a simple capability to adjust the mean frequency of the transmittable interval of the resonator circuit. The width of the resonance curve of the resonator circuit can, in particular, be adjusted by adjustment of its Q-factor. The resonator circuit may be regarded as a filter with non-linear attenuation, which in principle allows any desired level of dynamic compression to be achieved. Because the processing is carried out over a sufficiently narrow bandwidth, it is also possible to keep adequately low distortion which can occur as a result of excessive non-linearity.

The circuit arrangement may include a second-order resonator circuit, in which the attenuation rises non-linearly as the sound level rises. In the case of a passive implementation of the circuit arrangement, that is to say when using passive components (coil L, capacitor C, resistance R), it is possible to produce a stable circuit (in contrast to systems which require an active amplifier with feedback).

The resonator circuit may have a resistance which can be subjected to open-loop control (or closed-loop control) by means of the resonator control circuit. A resistance which can be subjected to closed-loop or open-loop control such as this is a simple circuit element, by means of which the functionality of closed-loop control of the Q-factor of the resonator circuit can be carried out with little complexity, as well as accurately and in a stable manner.

It should be noted that the resonator control circuit can be formed from a plurality of resonator control circuit elements, in which case one resonator control circuit element in each case provides open-loop or closed-loop control for the respective Q-factor of the resonator circuit associated with it.

The input signal may be provided between a first connection of the resistance and a first connection of the capacitance. The output signal may be produced between the first connection of the capacitance and a second connection of the capacitance. A second connection of the resistance may be coupled to a first connection of the inductance, and a second connection of the inductance may be coupled to a second connection of the capacitance.

The resonator control circuit may be designed in such a manner that it controls the Q-factor of the at least one resonator circuit on the basis of a Boltzmann function, in which the amplitude of the output signal is included as a parameter. If the parameters contained in it are chosen appropriately, a Boltzmann function is highly suitable for approximation of the sensitivity curve of the outer hair inner cells in the human inner ear. A particularly good description of this biological relationship can be provided by means of a second-order Boltzmann function. This makes it possible to approximate the sensitivity curve in the human ear, as is advantageous for applications of the circuit arrangement in the medical field (for example for a hearing aid).

The resonator control circuit may be designed in such a manner that it adjusts the Q-factor of the at least one resonator circuit as a function of the amplitude of the output signal, based on a sensitivity characteristic determined for a human ear. In order to simulate the sensitivity characteristic in the inner ear of a human being particularly well by means of a circuit arrangement according to the invention, a sensitivity characteristic (which has been determined, for example, experimentally or theoretically) of the human ear can be stored in the form of a file or table, such that it is accessible to the control circuit. In this case, the resonator control circuit can provide open-loop or closed-loop control for the Q-factor of the

at least one resonator circuit, in such a manner that the biological sensitivity characteristic stored in it is approximated.

The resonator control circuit may be designed in such a manner that it adjusts the Q-factor of the at least one resonator circuit to be less the higher the amplitude of the respective output signal element from the respective resonator circuit is.

The resonator control circuit may also be designed in such a manner that it adjusts the Q-factor of the at least one resonator circuit as a non-linear function of the amplitude of the respective output signal element. That is to say signal areas of high amplitude are attenuated to a more than proportional extent in comparison to signal areas of low amplitude. Thus, even if there is an extremely wide range of sound levels in an input signal, it is possible to achieve a compromise with a sufficiently narrow range in the output signal.

The resonator control circuit can be designed in such a manner that it adjusts the Q-factor of the at least one resonator circuit in such a manner that the amplitude of the respective output signal element is within a predetermined interval. For specific applications, it may be advantageous to keep the amplitude of an output signal element within a predetermined interval in all cases. By way of example, this may be important for data compression purposes, when a signal with a high degree of intensity fluctuation is intended to be detected with as few quantization levels as possible. In this case, the resonator control circuit can be designed in such a manner that it provides open-loop or closed-loop control for the Q-factor of the resonator circuit in such a manner that the respective output signal element is within the predetermined interval.

The circuit arrangement may have a plurality of series-connected resonator circuits, in which case an output from a respective upstream resonator circuit can

be produced as the input signal for the respective resonator circuit connected downstream from it.

As can be seen, according to this ~~particularly advantageous~~ refinement, a filter bank with a plurality of resonator circuits connected in series is provided, thus making it possible to extend the dynamic compression to an even wider dynamic range. In principle, a sufficiently high degree of dynamic compression (for example 60 db) can be achieved just with one filter stage (that is to say with one resonator circuit) with a very high Q-factor Q (for example Q = 1000, which is reduced to a Q-factor of Q = 1 for high levels). A circuit arrangement such as this has a very narrow bandwidth, however (for example 0.1% of the resonant frequency of the resonator circuit). According to the invention, a sufficiently high degree of dynamic compression (for example of 60 dB) can be achieved by cascading a plurality of filter stages (for example three series-connected filter stages) with a relatively low Q-factor Q (for example Q = 10, so that $Q^3 = 1000$). The individual Q-factors, which are not as high, of each of these filters result in the advantageous effect that, because of the increased bandwidth of the individual filters which results from the lower Q-factor, the filters cover a wider frequency range, and the impulse response by the filters is at the same time improved, that is to say the transient times of the system, on switching on and off, are considerably shorter.

As can be seen, the series-connected resonator circuits are coupled directly to one another in such a manner that the output voltage of an upstream resonator circuit is at the same time the input voltage to the resonator circuit connected downstream from it, and in such a manner that the output current (which is generally not zero during operation) from an upstream resonator circuit is at the same time the input current for the resonator circuit connected downstream from it. For this purpose, the circuit arrangement generally has no intermediate element between upstream and downstream resonator circuits. This can be achieved by means of a circuit arrangement in which the second connection of the coil of an

upstream resonator circuit is coupled to the first connection of the resistance in the resonator circuit which is connected downstream from the upstream resonator circuit.

Alternatively, as can be seen, the series-connected resonator circuits can be free of direct coupling, that is to say they can be decoupled from one another in a certain manner, in particular by the interposition of an intermediate element between the output of an upstream resonator circuit and the input of a downstream resonator circuit. This is preferably implemented in such a manner that the output voltage from an upstream resonator circuit is at the same time the input voltage for the resonator circuit connected downstream from it, and in such a manner that the output current from an upstream resonator circuit is equal to zero. The input current to the downstream resonator circuit essentially results from the impedance of this resonator circuit. In the case of a circuit arrangement such as this, an operational amplifier is preferably provided (as an impedance converter) as the intermediate element between an upstream resonator circuit and the resonator circuit connected downstream from it. A first input of the operational amplifier is coupled to the second connection of the coil of the upstream resonator circuit. A second input of the operational amplifier is fed back to an output of the operational amplifier, and is coupled to the first connection of the resistance in the resonator circuit which is connected downstream from the upstream resonator circuit.

In order to reduce the computation power, the Q-factor of all the series-connected resonator circuits can be set to be identical. In this case, the computation power which is required by the resonator control circuit is kept particularly low, since a common Q-factor is determined and set for all of the resonator circuits, that is to say all of the filter parameters are identical. If a circuit arrangement with a particularly high quality requirement is needed, then, alternatively, the Q-factor of the various series-connected resonator circuits can be set to be different for optimization purposes. In a circuit arrangement such as

this, the Q-factor of each of the series-connected resonator circuits is thus set individually.

The circuit arrangement preferably has a plurality of parallel-connected branches, with each branch having one resonator circuit or a plurality of series-connected resonator circuits. In this case, the Q-factor of each resonator circuit can be subjected to open-loop or closed-loop control by means of the respective resonator control circuit.

According to this ~~particularly advantageous development~~ embodiment of the invention, a plurality of parallel-connected branches of resonator circuits are clearly provided, in which case a plurality of resonator circuits may be connected in series in each branch.

The at least one resonator circuit in each branch is preferably designed in such a manner that it passes a respective frequency range of the input signal in such a manner that the branches together pass a cohesive frequency interval. The frequency range for which the human ear is sensitive is between about 20 Hz and 20 kHz. In order to cover this hearing frequency range, the frequency ranges of the signals which can be transmitted in different channels in the parallel arrangement of resonator circuits are in general different. The frequency range of signals which can be transmitted in one resonator circuit is a distribution curve around the resonant frequency, with a specific 3-D width. The resonant frequency can, as can be seen, be adjusted by adjustment of the values L, C of the resonator circuit, and the 3dB width can be adjusted by adjustment of the respective Q-factor. If the various frequency pass-bands of the various branches of resonator circuits are joined together, then this results in a preferably cohesive frequency interval, by means of which the sensitivity range of the human hearing, or of any other frequency range of interest, can be covered.

The frequency ranges which the various branches pass preferably at least partially overlap one another. This ensures that all of the frequencies are covered, and that the signal components from the individual branches can be combined.

The frequency range which a respective branch passes can preferably be predetermined by adjustment of the value of the capacitance and/or of the inductance of the at least one resonator circuit in that branch. This is based on the fact that the resonant frequency of a resonator circuit depends on the values of the inductance and of the capacitance.

The circuit arrangement according to the invention is preferably designed to process an acoustic signal as the input signal. In this case, the circuit arrangement according to the invention is suitable for use in a speech processing system. A system such as this may, for example, be based on pulsating neural networks which are instructed to reduce the dynamic range. Further fields of application are systems for sound processing and (audio) data compression, when signals with high amplitudes are intended to be recorded with as few quantization levels as possible. Furthermore, there are applications in the medical field, in particular as a hearing aid for patients with loud-noise deafness.

The circuit arrangement according to the invention can be produced using digital or analog circuit technology.

At least a part of the circuit arrangement, in particular the filters, the open-loop or closed-loop control functionality of the resonator control circuit, may be in the form of a computer program. The invention can be implemented by means of a computer program, that is to say in software, and by means of one or more specific electrical circuits, that is to say in hardware or in any desired hybrid form, that is to say by means of software components and hardware components.

A software implementation, in particular of the control circuit, can be produced, for example, using "C++". An implementation may be based on any desired processor or DSP (digital signal processor), or else or an FPGA module. FPGA ("Field Programmable Gate Array") is an integrated programmable circuit which in general has a large number of programmable cells on one chip.

According to another refinement of the invention, provision is made for the preferably linear filter bank to be in the form of a linear wave digital filter.

Furthermore, a plurality of high-pass filters may be provided, with each filter stage having at least one associated high-pass filter, and with one high-pass filter in each case being coupled to the output of a respective resonator circuit. These high-pass filters simulate the liquid coupling of the hair bundles of the senor cells in the inner ear to the oscillation of the basilar membrane.

The use of at least one high-pass filter per filter stage, coupled to the output of the respective resonator circuit, results in the otherwise relatively flat high-frequency filter flank of the filter bank being sharpened. At least some of the high-pass filters, preferably all of the high-pass filters, are preferably in the form of first-order high-pass filters. According to one development of the invention, the cut-off frequency of at least some of the first-order high-pass filters is chosen in such a manner that it corresponds to the frequency of the maximum sensitivity of a basilar membrane oscillation of an inner ear of a mammal.

According to another refinement of the invention, a plurality of rectifier circuits are provided, with one rectifier circuit in each case being associated with one of the filter stages and one high-pass filter, and being coupled to one output of a respective high-pass filter, preferably as well as a plurality of low-pass filters, with one low-pass filter in each case being associated with one rectifier circuit and being coupled to one output of a respective rectifier circuit. These refinements

result in a very good approximation to the formation of the receptor potential U_M in the human hearing system.

Furthermore, a plurality of activation circuits may be provided, with one activation circuit in each case being associated with one of the filter stages, and with each activation circuit being designed to amplify any rate of change of a signal supplied to the activation circuit, and for attenuation of components of the signal which is supplied to the activation circuit which components are essentially constant over time.

Furthermore, each activation circuit preferably has a vesicle pool circuit with a large number of vesicle circuits.

The signal-processing apparatus according to the invention, which has a circuit arrangement according to the invention, will be described in more detail in the following text. The refinements of the signal-processing apparatus also apply to the circuit arrangement, and vice versa.

In the signal-processing apparatus, the further-processing unit may be a speech-recognition device or a hearing aid.

One implementation of the further-processing unit in the form of a hearing aid is intended in particular for an application in which dynamic compression is carried out in order to compensate for disturbances in the volume awareness of those who are deaf. In those with disturbed hearing, the outer hair cells may be defective, as a result of which the increase in sensitivity fails at low sound levels. The hearing then, as can be seen, always operates with the sensitivity that it has at high sound levels. This leads to narrowing (recruitment) of the useable range of sound levels between the hearing threshold (very quiet) and the unpleasantness threshold (very loud). In order to compensate for this phenomenon, the circuit arrangement for the signal-processing apparatus according to the invention can be

used to carry out dynamic compression, which, as can be seen, compresses the wide sound level range from the acoustic environment to the range to be perceived by the patient.

The signal-processing apparatus may also form the input of a speech-recognition system, in particular in a pulsating neural network architecture.

The signal-processing apparatus may be designed as an analog or digital filter bank.

The following principles must clearly be maintained within the scope of the invention:

a) the input signal provided for the circuit arrangement is subjected to frequency analysis with non-linear dynamic compression;

b) a reduction in information was provided by "soft" half-wave rectification of the frequency channels that are formed, with a threshold value and saturation;

c) speech-relevant modulation frequencies are emphasized by simulation of the neural adaptation in the hearing system of a mammal, in particular of a human being.

~~Exemplary embodiments of the invention will be explained in more detail in the following text and are illustrated in the figures, in which:~~

~~Figure 1 shows a circuit arrangement according to one preferred exemplary embodiment of the invention;~~

~~Figure 2 shows a resonator circuit according to one exemplary embodiment of the invention;~~

~~Figure 3 shows an implementation of the resonator circuit as shown in Figure 2, in the form of a wave digital filter;~~

~~Figures 4 and 5 show diagrams to illustrate the functionality of the circuit arrangement shown in Figure 1;~~

~~Figure 6a shows a circuit arrangement element according to another exemplary embodiment of the invention;~~

~~Figure 6b shows an implementation of the resonator circuits illustrated in Figure 6a as wave digital filters;~~

~~Figure 7a shows a circuit arrangement element according to another exemplary embodiment of the invention;~~

~~Figure 7b shows an implementation of the resonator circuits illustrated in Figure 7a as wave digital filters;~~

~~Figure 8 shows a block diagram of a speech recognition system according to one exemplary embodiment of the invention;~~

~~Figure 9 shows a circuit diagram of a linear filter bank and of a plurality of resonator circuits associated with individual filter stages in the filter bank according to one exemplary embodiment of the invention;~~

~~Figure 10 shows a diagram illustrating an excitation pattern for a non-linear basilar membrane model for a 1 kHz tone;~~

~~Figure 11 shows a circuit diagram of a circuit element which is in each case connected in series with a respective resonator circuit in order to form the circuit arrangement according to one exemplary embodiment of the invention,~~

~~Figure 12 shows an illustration of modeled nerve action potentials, and~~

~~Figure 13 shows a diagram illustrating different speech-recognition rates for a speech-recognition system according to one exemplary embodiment of the invention, compared with a speech-recognition system according to the prior art.~~

~~Figure 8 shows a speech-recognition system 800 according to one exemplary embodiment of the invention.~~

Figure 8 illustrates a speech-recognition system 800 according to one exemplary embodiment of the invention.

For the purposes of the speech recognition system, a feature-extraction system 801 is used which extracts from the analog speech signals applied to it the features which are used for the actual automatic speech-recognition process (illustrated in Figure 8 by a speech-recognition block 802).

The feature-extraction system 801 has, in particular, components for initial filtering, a filter bank and components for non-linear feature extraction.

As can be seen, the feature-extraction system 801 simulates the signal-processing strategy and the signal-processing structure of the human hearing system. This is scaled in physical units analogously to the signal-processing system of the human hearing system.

The feature-extraction system 801 is supplied with an input signal 803 in analog form, as a sound pressure signal (measured in Pascal).

A first component 804 of the feature-extraction system 801 forms a model of the auditory channel which, however, is optional and is ignored in the preferred implementation.

The signal 805, which is formed by the model of the auditory channel 804 from the input signal 803, is supplied to a middle-ear model component 806.

As illustrated in Figure 9, the middle-ear model component 806 has an ideal spring component 901 and an ideal damping component 902 connected in parallel (implemented in the form of an electrical core, that is to say an inductance, for the ideal spring and in the form of a resistance as the damping component). The middle-ear model component 906 is designed in such a manner that the speech-relevant region of the spectrum of the input signal is emphasized, that is to say it is amplified. By analogy with the human hearing system, the signal 807 which is produced by the middle-ear model component 806 is supplied to an inner-ear model component 808.

The signals 809 which are formed by the inner-ear model component 808 can optionally be used directly as extracted speech-recognition features for speech recognition, or can be supplied to a sensor-cell model component 810, which will be described in more detail in the following text. The signals 811 which are produced by the sensor-cell model component 810 can likewise be used directly as feature components for automatic speech recognition purposes, or may be processed further and, in the course of this further processing, may be supplied to a synaptic-model component 812, which simulates the synaptic mechanism of the human hearing system. The signals 813 which are formed by the synaptic-model component 812 are likewise used according to the invention as features for automatic speech recognition purposes.

In this context, it should be noted that, on the basis of alternative refinements of the invention, one, two all or three of the feature signals 809, 811, 813 described above may be used optionally for speech recognition purposes.

Within the human hearing system, the sound pressure which arrives at the tympanic membrane (the tympanic membrane has a surface area of $A_{ed} = 55 \times 10^{-6}\ m^2$) is converted to a mechanical deflection of the middle-ear bones (quoted in m).

As has been described above, the middle-ear model component 806 has a low-pass filter, according to this exemplary embodiment a first-order low-pass filter, preferably with a low-pass filter cut-off frequency of 1 kHz.

The spring constant that is simulated by means of the inductance 901 is about 1500 N/m, and the damping component which is represented by means of the resistance 902 is designed to be 0.25 Ns/m.

The signal 807 which is produced by the middle-ear model component 806 is injected into the inner-ear model component 808, as is illustrated in detail in Figure 9. The inner-ear model component 808 is in the form of a filter bank 903, according to this exemplary embodiment in the form of a linear wave digital filter model.

The filter bank 903 has a large number of filter stages 904, 905, 906, as well as a terminating resistance 907.

Each filter stage 904, 905, 906 is formed by an inductance 904a, 905a, 906a, a resistance 904b, 905b, 906b and a capacitance 904c, 905c, 906c, connected in series.

The speed of basilar membrane oscillation in the human inner ear in each case corresponds to the current in one filter stage 904, 905, 906. The deflection of the basilar membrane can thus be calculated by integration of the speed. In order to avoid numerical problems during the integration process, it is, however, worthwhile calculating the deflection in a different way:

In the case of a spring, as represented by the capacitances 904c, 905c, 906c, the instantaneous deflection x can be calculated as the product of the spring force and the spring constant. This deflection x forms the input, that is to say the input signal, for the dynamic-compression stages 101, according to this exemplary embodiment of the invention, in each case two resonator circuits 101 connected in series, as will be explained in more detail in the following text.

The resonator circuits 101, as can be seen, form compression stages using second-order resonators, as will be explained in more detail in the following text.

In one alternative preferred embodiment, four resonator circuits 101 are provided connected in series for each filter stage 904, 905, 906.

As will likewise be explained in more detail in the following text, the Q-factor of the resonator circuits 101 is in each case modulated in a range from unity up to a maximum Q-factor value Q_{max} (depending on the position in the person's inner ear) instantaneously as a function of the output signal from each filter stage 904, 905, 906.

The design of the resonator circuits 101 will be explained in more detail in the following text with reference to Figure 1.

Circuit arrangement 100 contains a large number of resonator circuits 101, each of which has a capacitance and an inductance (not ~~shown~~illustrated in Figure 1), as well as an input to which an input signal can be applied, and an output at

which an output signal can be produced. Three of the resonator circuits 101 are in each case connected in series along a respective row of the arrangement, which is in the form of a matrix, so that a respective output of an upstream resonator circuit 101 is coupled to a respective input of a resonator circuit 101 connected downstream from it. The values of the inductance and of the capacitance in the resonator circuit 101 in one row are in each case chosen in such a manner that the respective row can transmit a signal of a corresponding frequency interval in a region around the resonant frequency of the resonator circuit 101 for that row. The resonator circuits 101 for different rows each have different L, C values, so that, overall, the individual rows or branches of resonator circuits 101 cover a cohesive frequency interval, which corresponds to the sensitivity range of the human hearing (approximately 20 Hz to 20 kHz).

A resonator control circuit 111 is coupled to all of the resonator circuits 101 in a communication link, that is to say the control circuit 111 is coupled to all of the resonator circuits. The Q-factor of each individual one of the resonator circuits 101 can be adjusted by means of the control circuit 111 in order to provide open-loop or closed-loop control for the Q-factor of the resonator circuits 101, with the control circuit 111 being designed in such a manner that it adjusts the Q-factor of the resonator circuits 101 as a function of the amplitude of an output signal from the last resonator circuit 101 in a respective row. By way of example, the Q-factor of the resonator circuits R_{11}, R_{12}, R_{13} is adjusted by means of the resonator control circuit 111 on the basis of the amplitude of a signal at the output of the resonator circuit R_{13}.

Figure 1 also ~~shows~~illustrates a sound source 103 which emits an acoustic signal as a global input signal 102. This is applied to the inputs of the resonator circuits 101 (R_{11}, R_{21}, …, R_{k1}, …, R_{n1}) in the first column of resonator circuits 101.

The resonator circuit 101 R_{11} arranged in the first row and the first column of resonator circuits will be analyzed in the following text. The global input

signal 102 from the sound source 103 is applied to this at one input. The resonator circuit 101 R_{11} passes a frequency component of the global input signal 102 which is dependent on the values L and C associated with it, and this frequency component is produced as the first local output signal 104 at one output of the resonator circuit R_{11}. Furthermore, the amplitude of the global input signal 102 is changed on the basis of the functionality of the resonator circuit 101 R_{11}, depending on its (present) Q-factor Q. The Q-factor Q of the resonator circuit 101 R_{11} is controlled by means of a resistance (not ~~shown~~illustrated in Figure 1) in the resonator circuit 101 R_{11}, with the control circuit 111 providing an appropriate control signal for this controllable resistance, by which means the impedance is set to a predetermined value. This results in adjustment of the Q-factor of the resonator circuit 101, so that an input signal is attenuated to a greater or lesser extent, corresponding to this value of the Q-factor, in a subsequent processing cycle. Since the circuit arrangement 100 is designed for dynamic compression of the global input signal 102, signal ranges of high amplitude are, as can be seen, attenuated to a greater extent than signal ranges of low amplitude.

The first local output signal 104 is provided as the first local input signal 105 to the resonator circuit 101 R_{12} connected downstream from the resonator circuit 101 R_{11}. The first local input signal 105 passes through the resonator circuit 101 R_{12}, with the second local output signal 106 being produced at one output. The second local output signal 106 is used as the second local input signal 107 for the resonator circuit 101 R_{13} which is connected downstream from the resonator circuit 101 R_{12}. A third local output signal 108 is produced at its output 108. This is assembled (added) together with the output signals (which are each related to a separate frequency interval) from the respective last resonator circuit 101 (R_{13}, R_{23}, ..., R_{k3}, …, R_{n3}) arranged in a row to form a global output signal 109.

In each of the resonator circuits 101 in a respective row of resonator circuits (R_{k1}, R_{k2}, R_{k3}), the Q-factor of all of the resonator circuits 101 in the row is

controlled by means of the resonator control circuit 111 on the basis of the amplitude of the output signal at the output of the respective last resonator circuit (in the k-th row resonator circuit R_{k3}).

The assembled global output signal 109 is thus subjected to dynamic compression with respect to the global input signal 102.

The resonator circuit 101 from Figure 1 will be described in the following text with reference to Figure 2.

An input signal 200 is symbolized as the voltage source U in Figure 2. Furthermore, an output signal 204 is symbolized as the voltage U_c. The input signal 200 is produced between a first connection of a resistance 203 and a first connection of a capacitance 201. The output signal 204 is produced between the first connection of the capacitance 201 and a second connection of the capacitance 201. Furthermore, a second connection of the controllable resistance 203 is coupled to a first connection of an inductance 202, and a second connection of the inductance 202 is coupled to the second connection of the capacitance 201.

The value of the resistance R 203 can be adjusted by means of the control circuit 111. The resonator circuit 101 from Figure 2 thus clearly represents a filter whose attenuation is controllable.

In the circuit arrangement 100 according to the invention, three (or in general N) resonator circuits 101 are connected in series as filter elements without any feedback in each row. The time-dependent output signal $U_C(t)$, where t is the time, from an upstream filter in each case defines the input signal U 200 for the filter which is connected downstream from the upstream filter.

The resistance R 203 can be varied as a non-linear function of the output voltage $U_c(t)$ (closed-loop control), as a function of $U_C(t)$ of the respective

upstream filter (open-loop control), or else simultaneously for all of the filters as a function of $U_C(t)$ of the last filter stage in a row.

The following text describes the calculation rule which is used as the basis, according to the described exemplary embodiment, for adjustment of the value R of a respective resistance R 203.

First of all, a Q-factor Q to be set is calculated for this purpose.

According to the described exemplary embodiment, the Q-factor Q of the filter is damped according to a Boltzmann function:

$$Q(t) = (Q_0 - Q_{min})(1 - [\frac{2}{1 + \exp\{-SAT\,|\,U_C(t)\,|\}} - 1]) + Q_{min} \qquad (1)$$

In the equation (1), Q(t) is the relationship between the Q-factor Q and the time t. Q_0 is a predeterminable maximum Q-factor of the resonator circuit 101 (for example $Q_0 = 10$). Q_{min} is a predeterminable minimum Q-factor of the resonator circuit (for example $Q_{min} = 1$). SAT is a predeterminable saturation threshold, that is to say a parameter by means of which the time dependency of the Q-factor can clearly be adjusted (for example SAT = 1).

The Boltzmann function (1) approximates the sensitivity curve of the outer hair sensing cells in the inner ear. If required, the function can be replaced by a second-order Boltzmann function, which allows even more accurate matching, with the introduction of a further parameter. Equation (1) uses a simple first-order Boltzmann function, since it has only one free parameter (specifically SAT) and can thus be processed with little numerical complexity.

The value to be set for the non-linear resistance can be calculated from the Q-factor Q of the filter using:

$$R(t) = \sqrt{L/C}\,Q(t) \qquad (2)$$

The time-dependent value of the resistance R(t) thus depends on the value of the inductance L and of the capacitance C, as well as on the time-dependent Q factor Q(t).

Equations (1) and (2), as can be seen, form the control rule for adjustment of the value R of the resistance 203 by means of the control circuit 111.

The filter which is formed by the resonator circuit 101 illustrated in Figure 2 is linear for very low amplitudes $U_C(t)$ (where $Q \rightarrow Q_0$ for $U_C(t) \rightarrow 0$). It is likewise approximately linear for very high amplitudes $U_c(t)$ ($Q \rightarrow Q_{min}$ for $U_C(t) \rightarrow \infty$). The dynamic compression K is carried out in the region of the saturation threshold (SAT) and is $K = Q_0/Q_{min}$. If there are N = 4 series-connected filter stages (although only three filter stages are provided in Figure 1 by means of three resonator circuits in one row), and if the values are $Q_0 = 10$ and $Q_{min} = 1$, this allows major compression to be achieved, by 80dB ($K_N = (Q_0/Q_{min})^N$).

In order to cover the entire human hearing range, a filter bank is provided having resonant frequencies in the range from approximately 20 Hz to approximately 20 kHz, and this is typically achieved by the use of 50 to 100 rows of resonator circuits 101 (that is to say n = 50 to n = 100). According to the described exemplary embodiment, the value of the inductance is set to L = 1 H. The respective value C is then calculated for each row of resonator circuits 101 on the basis of the filter frequency f_0 that is covered by this row, from the resonator frequency of the corresponding LC element:

$$C = (4\pi^2 f^2 L)^{-1} \qquad (3)$$

It should be noted that the non-linear Q-factor for each filter frequency f_0, that is to say for each row of resonator circuits 101, is calculated independently. With reference to Figure 1, this means that each row of oscillator circuits 101 has an associated corresponding filter frequency f_0 for which the value of the Q-factor Q(t) is calculated.

A wave digital filter 300, as one implementation of the resonator stage 101 ~~shown~~illustrated in Figure 2, will be described in the following text with reference to Figure 3.

A wave digital filter represents one class of digital filters with particularly advantageous characteristics. These are traditional filters modeled from conventional components from information technology, and are operated with the aid of modern integrated digital circuits. On the basis of the technology of a wave digital filter, an analog model can, as can be seen, be implemented digitally (for example using a computer).

The components of the wave digital filter 300 in Figure 3 are, as can be seen, associated in the following text with the components of the resonator circuit 101 from Figure 2, and the corresponding variables are defined.

A first block 301 of the wave digital filter 300 contains a reflection-free serial coupler with the impedances R11 and R13. R11, as can be seen, represents the controllable resistance R 203, with respect to a reference impedance. R21 represents a corrected impedance (impedance) of the coil L 202 with respect to a base frequency. A second block 302 contains a parallel coupler, which reflects the parallel connection of the capacitance 201, with the admittances G21, G22, G23 being illustrated in the second block. G21 is an input admittance of the

second block (G12=1/R13) 302, G23 is an output admittance of the second block 302. The admittance G22 is used to model the impedance of the capacitance C 201. A third block 303 represents a store or a filter register for the capacitance 201, and a fourth block 304 represents a store or a filter register for the coil 202.

The variables illustrated in Figure 3 will be defined in the following text. The parameters for a wave digital filter, for each filter frequency are:

$$R11=R/R\_B \quad (4)$$
$$R12=2\pi\ F\_B\ L/(R\_B\ \tan[\pi\ F\_B/f\_s]) \quad (5)$$
$$R13=R11+R12 \quad (6)$$
$$G21=R13^{-1} \quad (7)$$
$$G22=2\pi\ F\_B\ C\ R\_B/\tan(\pi\ F\_B/f\_s) \quad (8)$$
$$G23=G21+G22 \quad (9)$$

In this case, R is the resistance 203 and R_B is a predeterminable reference impedance. F_B is a predeterminable reference frequency. The values R_B and F_B are used for scaling. Since the implementation based on the described exemplary embodiment is carried out using double-precision float variables, this normalization is not relevant, although it is if integer arithmetic is used. L is the inductance of the coil 202. The value f_s is a sampling frequency of the sampled time signal. The variables R11, R12, R13 are resistances, while in contrast to the variables G21, G22 and G23 are admittances, that is to say inverse resistances.

The filter coefficients g1, g2 are:

$$g1=R11/R13 \quad (10)$$
$$g2=G21/G23 \quad (11)$$

The initial values of the filter registers Z1 (fourth block 304) and Z2 (third block 303) are initialized by zero.

The signals at the individual ports can be calculated successively. For the "forward wave" of the signal, that is to say, as can be seen, the coefficients on the arrows which are oriented to the right in Figure 3, then:

b13=-(U+Z1) (12)

b20=-g2(Z2-b13) (13)

B23=b20+Z2 (14)

The variable U in equation (12) is the input signal 200.

For the "backward wave", that is to say, as can be seen, the arrows which are oriented to the left in Figure 3, the coefficients are:

b22=b20+b23 (15)

b21=b22+Z2-b13 (16)

a0=b21-b13 (17

b11=U-g1 a0 (18)

b12=-(b11+b21) (19)

The output signal U_c 204 is then calculated to be:

U_c=(b22+Z2[sec])/2 (20)

The filter registers (blocks 303, 304) are updated as follows:

1=-b12 (21)

Z2=b22 (22)

The output signal U_c 204 is passed as the input signal U 200 to the filter stage 101 downstream from the filter stage 101 under consideration. The

Q-factor to be set for the series-connected filters 101 is determined again using equation (1) on the basis of the output signal U_c 204 from the last filter stage 101 in one row of filter stages 101. The value of the impedance R which governs the attenuation is calculated in accordance with equation (2) from the value of the Q-factor Q determined in this way. The filter impedances (R11, R12, R13, G21, G22, G23) and filter coefficients (g1, g2) are calculated again using equations (4) to (11) using the amended value of the resistance R 203. After this ~~step~~process, the output signal is calculated for the next time slice. In other words, the time spectrum can be broken down into a plurality of time slices, which are numerically calculated successively.

A diagram 400 will be explained in the following text with reference to Figure 4, in which the functionality of the circuit arrangement according to the invention is ~~shown~~illustrated on the basis of one preferred exemplary embodiment of the invention. The diagram 400 relates to a circuit arrangement having N=4 series-connected resonator circuits. The maximum Q value is assumed to be Q_0=10, and the minimum Q-value is assumed to be Q_{min}=1.

The frequency of a signal, normalized with respect to a reference frequency f_0, is plotted on a logarithmic scale along the abscissa 401 of the diagram 400. The reaction of the system to an input signal of a specific intensity is ~~shown~~illustrated on a logarithmic scale along the ordinate 402. First to eighth curves 403 to 410 represent the frequency response (that is to say in this case the respective value of the maximum amplitude of the filter output) of the circuit arrangement according to the invention for different signal amplitudes (with respect to a reference amplitude). The first curve 403 corresponds to an amplitude of $1x10^{-9}$, the second curve 404 corresponds to an amplitude of $1x10^{-4}$, the third curve 405 corresponds to an amplitude of $1x10^{-3}$, the fourth curve 406 corresponds to an amplitude of $1x10^{-2}$, the fifth curve 407 corresponds to an amplitude of $1x10^{-1}$, the sixth curve 408 corresponds to an amplitude of $1x10^{0}$, the seventh curve 409 corresponds to an amplitude of $1x10^{6}$, and the eighth curve 410

corresponds to an amplitude of $1x10^6$. Furthermore, the input signal is assumed to be a sinusoidal oscillation, which is windowed using a $\cos^2$ window. Curves 403 to 410 are obtained for an entire filter bank ~~comprising~~having N=4 series-connected resonator circuits without feedback.

First of all, the diagram 400 ~~shows~~illustrates that the attenuation of the input signal becomes greater the higher the signal intensity or signal amplitude. If the amplitudes are very low, the filters are linear and the resonant peak is approximately 80dB. In response the filter bank decreases very sharply towards high frequencies, because the filters are in the form of low-pass filters (see Figure 2). The high-frequency response of the filters falls at approximately 6dB per octave, because the filter parameters are scaled by f_0. The curves in Figure 4 simulate the highly asymmetric frequency selectivity of the human hearing, to a good approximation.

The relationship between the amplitudes of the input signal and of the output signal of a circuit arrangement according to the invention will be described in the following text with reference to Figure 5.

The sound pressure level A_{in} is plotted on a logarithmic scale in dB along the abscissa in the diagram 500, with respect to a sound pressure of the reference variable 20µPa. The strength of the output signal A_{out} is plotted in dB in arbitrary units along the ordinate 502. Curves 503 to 507 ~~show~~ illustrate (for different scenarios) the growth function of a filter cascade ~~comprising~~having four resonator circuits (N = 4 filters connected in series) at the resonant frequency f_0. The minimum Q-factor is assumed to be $Q_{min} = 1$.

The first curve 501 ~~shows~~illustrates a linear growth function. The second curve 504 ~~shows~~illustrates a growth function of the inner ear, that is to say the speed of the basilar membrane with respect to the sound pressure measured in front of the tympanic membrane. The data for the second curve 504 is taken from [2].

A third curve 505 ~~shows~~illustrates the curved profile for a Q-factor of Q = 2, a fourth curve 506 ~~shows~~illustrates the profile for Q = 4, and a fifth curve 507 ~~shows~~illustrates the profile for Q = 10.

As can be seen, Figure 5 ~~shows~~illustrates the growth function of a filter output for $f = f_0$ with the filter variable Q as a parameter. The growth functions are approximately linear at very low and very high amplitudes. The wide compression range (in particular for a high Q) is conspicuous, extending over more than four decades. The wide dynamic range of the input signal (100 dB) is compressed to 40 dB (for Q = 10). Quiet signals are "amplified" on a frequency-specific basis because of the resonant peak. The growth function provides a very good simulation of the oscillation responses measured on a living hearing system (see the curve 504). The circuit arrangement according to the invention therefore provides an approximate technical model of the non-linear preprocessing in the inner ear.

A circuit arrangement 600 according to another preferred exemplary embodiment of the invention will be described in the following text with reference to Figure 6A.

The circuit arrangement 600 is formed from a first resonator circuit 601 and a second resonator circuit 602, each of which is designed in the same way as the resonator circuit 101 illustrated in Figure 2. The resonator circuit 602 is connected downstream from the first resonator circuit 601.

As can be seen, the circuit arrangement 600 can be regarded as a directly coupled implementation of two (N = 2) series-connected resonator circuits 601, 602.

As is ~~shown~~illustrated in Figure 6A, the second connection of the coil 202 of the upstream resonator circuit 601 is coupled to the first connection of the resistance 203 in the downstream second resonator circuit 602.

According to the exemplary embodiment of resonator circuits 601, 602, which are coupled directly to one another, as ~~shown~~illustrated in Figure 6a, the output voltage U_{C1} of the upstream resonator circuit 601 is equal to the input voltage to the following resonator circuit 602. Furthermore, the output current from the first resonator circuit 601 is equal to the input current to the second resonator circuit 602.

It should be noted that the values of the impedances R1 and R2, of the inductances L1 and L2 and of the capacitances C1 and C2 of the resonator circuits 601, 602 may differ from one another, and/or may be set/controlled differently.

One implementation of the resonator circuits 601, 602 illustrated in Figure 6a will be described as a wave digital filter 650 in the following text with reference to Figure 6B.

As can be seen, the wave digital filter 650 is formed from a first component 651, which represents the first resonator circuit 601, and from a second component 652, which represents the second resonator circuit 602. The two components 651, 652 are directly coupled to one another in the manner illustrated in Figure 6b, corresponding to the coupled configuration of the resonator circuits 601, 602 ~~shown~~illustrated in Figure 6a. The internal design of each of the components 651, 652 corresponds essentially to that of the wave digital filter 300 ~~shown~~illustrated in Figure 3.

A circuit arrangement 700 according to yet another exemplary embodiment of the invention will be described in the following text with reference to Figure 7A.

The circuit arrangement 700 is formed from a first resonator circuit 701 and a second resonator circuit 702, which are connected in series. As can be seen, the resonator circuits 701, 702 are connected in series in a configuration such that they are decoupled from one another, that is to say an intermediate element is connected between the resonator circuits 701 and 702.

Each of the resonator circuits 701, 702 is designed essentially in the same way as the resonator circuit 101 illustrated in Figure 2. Furthermore, an operational amplifier 703 is connected between the first resonator circuit 701 and the second resonator circuit 702, with a non-inverting input 703a of the operational amplifier 703 being coupled to the second connection of the coil 202 in the upstream first resonator circuit 701. Furthermore, an inverted input 703b of the operational amplifier 703 is fed back to its output 703c, and is coupled to the first connection of the resistance 203 in the second resonator circuit 702, which is connected downstream from the first resonator circuit 701.

On the basis of this configuration, the output voltage from the upstream resonator circuit 701 U_{C1} 204 is equal to the input voltage of the second resonator circuit 702, which is connected downstream from the first resonator circuit 701. The output current from each resonator circuit is equal to zero. The input current to the second resonator circuit 702 which is connected downstream from the upstream first resonator circuit 701 is governed by the impedance of the downstream second resonator circuit 702. As is ~~shown~~illustrated in Figure 7a, these situations can be implemented using analog technology by means of an impedance converter, which applies the output voltage from the upstream resonator circuit 701 to the input of the downstream resonator circuit 702.

A wave digital filter 750 will be described, as an implementation of the circuit arrangement 700 ~~shown~~illustrated in Figure 7A, in the following text with reference to Figure 7B.

The wave digital filter 750 is subdivided into a first component 751 and a second component 752, with the first component 751 representing the first resonator circuit 701, and with the second component 752 representing the second resonator circuit 702. As can be seen, the two components 751, 752 are coupled to one another by virtue of the functionality of the operational amplifier 703. The internal design of each of the components 751, 752 corresponds essentially to the configuration ~~shown~~illustrated in Figure 3. The input signal to the first component 751 is U, and the input signal to the second component 752 is U_{C1}.

The combination of the linear filter bank 808 with the non-linear compression stages 101 simulates, according to the invention, the non-linear oscillation response of the inner ear of mammals very well, as has been explained above in conjunction with Figure 5.

In particular, a high degree of dynamic compression is achieved for sound levels in the range from 0 dB_{SPL} to 120 dB_{SPL} to a range from 1 nm to 100 nm (this corresponds approximately to 40 dB) (see Figure 10), and this is of major importance for the further processing carried out in the course of feature extraction and speech recognition.

The diagram 1000 in Figure 10 illustrates the cochlea position along the abscissa 1001, and the deflection of the basilar membrane which occurs at each respective cochlea position, along the ordinate 1002. As can be seen, the diagram 1000 thus represents an excitation pattern (RMS value) of the non-linear basilar membrane model for a 1 kHz tone. The illustrated curves 1003, 1004, 1005, 1006, 1007, 1008, 1009 are very highly compressed at the point 1010 with the greatest sensitivity (illustrated as a dashed line in Figure 10) normally at the position of 21 mm with respect to the null position of the cochlea. Furthermore, Figure 10 ~~shows~~illustrates an excitation threshold 1011, above which the human hearing system perceives a signal deflection.

The basilar membrane signal x_{BM1} which is produced by the respective resonator circuits 101 at the end of each series circuit is supplied to a respective filter-output circuit 1100, as is illustrated in Figure 11.

Each filter-output circuit 1100 has a high-pass filter 1101, a rectifier circuit 1102 coupled downstream from it on the output side, a low-pass filter 1103 coupled downstream from it in the signal flow direction, an activation circuit 1104 coupled downstream from this in the signal flow direction, as well as a vesicle pool circuit 1105 and a neurotransmitter circuit 1106.

The respective basilar membrane signal x_{BM1}, x_{BMi}, …, x_{BMn} is high-pass filtered and scaled by means of the high-pass filter 1101, which has a capacitance 1107 and a resistance 1108, so that only the respective speech-relevant dynamic range is extracted by means of a second-order Boltzmann function. The still relatively flat flank of the filter curves of the inner ear, as is illustrated in Figure 10, is sharpened somewhat by means of the first-order high-pass filter 1101.

According to this exemplary embodiment of the invention, the cut-off frequency of the high-pass filter 1101 corresponds to the frequency of maximum sensitivity of the basilar membrane oscillation.

The asymmetry of the Boltzmann function that is used according to the invention results in rectification of the signal (produced according to this exemplary embodiment by means of the rectifier circuit 1102, which is low-pass-filtered in the next stage by means of the low-pass filter 1103, so that a receptor-potential signal U_M is produced at the output of the low-pass filter 1103.

According to this exemplary embodiment of the invention, the resistance 1109 in the low-pass filter 1103 has an admittance of $g_M = 60$ nS, by analogy with the cell membrane, and the capacitance 1110 of the low-pass filter 1103 has a capacitance of $C_M = 12$ pF, simulating the cell membrane.

The activation of the respective cell, simulated according to the invention by the vesicle pool circuit 1105 and the neurotransmitter circuit 1106, is calculated from the receptor potential U_M by means of a Boltzmann function.

The low-pass filtering and the rectification, as described above, have a number of effects:

a) at low signal frequencies, there is one and only one maximum excitation of the sensor cells, per cycle of the acoustic stimulus,

b) acoustic signals in the frequency range above the cut-off frequency of the inner hair cells lead to activation on the basis of their envelope curve, and

c) the sensitivity and saturation of the Boltzmann function result in focusing of the sound processing on speech-relevant information.

In other words, the further processing of the receptor-potential signal U_M is carried out by the vesicle pool circuit 1005 and the neurotransmitter circuit 1106 being designed such that changes over time in the sound signal (that is to say in the input signal) are emphasized, and signal components of the input signal which remain the same, and are essentially constant over time, are ignored (adapted).

This results in effective suppression of stationary signals (for example interference noise).

According to this exemplary embodiment of the invention, the adaptation is modeled by means of the vesicle pool circuit 1105, with the simulated vesicle pool being filled continuously (but slowly) to its nominal value. A neurotransmitter current (according to this exemplary embodiment of the invention at a rate of 28000/s) is produced by the vesicle pool circuit 1105 relative to the instantaneous

vesicle pool size and a probability which is derived using a Boltzmann function from the membrane potential of the inner hair cell.

In the case of large-amplitude sound signals, the majority of the vesicle pool is dissipated, so that subsequent signal components generate only a small signal, that is to say a signal with a small amplitude.

In phases during which the input signal has a small amplitude applied to it, the vesicle pool is regenerated again. In other words, this means that the vesicle pool circuit 1105 simulates the functionality described above, and is designed in such a way that two time constants are provided, specifically a first time constant of $\tau_1 = 140$ milliseconds and a second time constant of $\tau_2 = 3$ ms.

The neurotransmitter current flows into the "synaptic gap", where it is dissipated with a time constant $\tau_3 = 1$ ms by means of the neurotransmitter circuit 1106 which, according to the invention, is simulated by the neurotransmitter.

In addition to the vesicle function from the vesicle pool, a further neurotransmitter current is produced, which depends only on the membrane potential of the inner hair cell, so that the chosen model is based on an infinite vesicle pool size, and a rate of 9000/s.

The two neurotransmitter currents allow adequate coding of stationary and transient sound signals, that is to say adequate coding of nerve-action potentials.

The vesicle pool 1105 can be modeled not only continuously but also ~~comprising~~having discrete vesicles. In the case of discrete modeling, the neurotransmitter current is in the form of a stochastic process. This procedure is chosen in order to code the switching signal into discrete nerve-action potentials.

A nerve-action potential is illustrated in a nerve-action potential diagram 1200 ~~shown~~illustrated in Figure 12, and is triggered when the concentration of the respective neurotransmitter in the synaptic gap exceeds a predetermined threshold value, according to this exemplary embodiment 1.0 vesicles.

Figure 12 ~~shows~~illustrates the modeled nerve-action potentials that are produced for stimulation with a synthetic vocal "e"~~.~~.

If there are two formant frequencies, of the vocal "e", this results in excitations~~.~~. Furthermore, this is achieved on the time structure (in particular for the second format) which is modulated with the fundamental speech frequency (100 Hz corresponding to 10 ms)~~.~~.

One highly advantageous characteristic of the feature-extraction unit 801 according to the invention is that it can be evaluated and optimized on the basis of the achievable recognition performance in the course of a speech-recognition method in an automatic speech-recognition system~~.~~.

The diagram 1300 in Figure 13 illustrates the speech-recognition performance of a conventional speech recognition method based on Fast Fourier Transformation (word error rate curve 1301) with different processing stages according to the invention (only inner-ear model component), word error rate curve 1302 and sensor cell word error rate curve 1303 interference noise (plotted along the abscissa).

The respectively achieved word error rate is illustrated along the ordinate 1305 in Figure 13~~.~~.

As is ~~shown~~illustrated in Figure 13, without the existence of interference noise, the recognition performance of the conventional method based on Fast Fourier Transformation is of higher quality, due in particular to the maturity of the

algorithms, which have been developed over several years, but the robustness of the features provided according to the invention is obvious as the interference noise rises~~.~~.

Although specific embodiments have been illustrated and described herein, it will be appreciated by those of ordinary skill in the art that a variety of alternate and/or equivalent implementations may be substituted for the specific embodiments shown and described without departing from the scope of the present invention. This application is intended to cover any adaptations or variations of the specific embodiments discussed herein. Therefore, it is intended that this invention be limited only by the claims and the equivalents thereof.

~~The following publications are cited in this document:~~

~~[1] E.G. Schukat-Talamazzini, Automatische Spracherkennung [Automatic speech recognition], Friedrich Vieweg & Sohn Verlagsgesellschaft, Braunschweig Wiesbaden, ISBN 3-528-05492-1, Chapters 1 to 3, 1995;~~

~~[2] S. Seneff, A Computational Model for the Perforal Auditory System: Application to Speech Recognition Research, Proceedings of IEEE ICASSP 1986, Tokyo, pages 1983 to 1986, Tokyo, 1986;~~

~~[3] M. Hunke and T. Holton, Training Machine Classifiers to Match the performance of Human Listeners in a Natural Vowel Classification Task, ICSSLP, pages 574 to 577, 1996;~~

~~[4] S. Sandhu and O. Ghitza, A Comparative Study of MEL Cepstra and EIH for Phone Classification under Adverse Conditions, IEEE, pages 409 to 412, 1995;~~

~~[5] C. Sumner et al., A Revised Model of the Inner Hair Cell and Auditory Nerve Complex, Journal of Acoustic Society of America, Vol. 111, pages 2178 to 2188, May 2002;~~

~~[6] T. Dau, Modell der effektiven Signalverarbeitung im Gehör, Einblicke [Model of the effective signal processing in the hearing, Insights], No. 29, pages 16 to 18, April 1999;~~

~~[7] V. Hohmann, Signalverarbeitung in digitalen Hörgeräten, Einblicke [Signal processing in digital hearing aids, Insights], No. 33, pages 24 to 26, June 2001;~~

~~[8] H. W. Strube, A Computationally Efficient Basilar Membrane Model, Acoustica, Vol. 58, pages 207 to 214, 1985;~~

~~[9] M. A. Ruggero et al., Mechanical Basis of Frequency Tuning and Neural Excitation at the Base of the Cochlea:~~

~~Comparison of Basilar Membrane Vibration and Auditory Nerve-Fiber Responses in Chinchilla, Proceedings of National Academy of Science USA, Vol. 97, No. 22, pages 11744 to 11750, October 2000;~~

~~[10] P. Dallos et al., The Cochlea, ISBN 0387944494, Springer-Verlag, Chapter 6, pages 318 to 385, 1998;~~

~~[11] DE 691 31 095 T2.~~

~~List of Reference Symbols~~

~~100 Circuit arrangement~~
~~101 Resonator circuits~~
~~102 Global input signal~~
~~103 (Sound) signal source~~
~~104 First local output signal~~
~~105 First local input signal~~
~~106 Second local input signal~~
~~107 Second local input signal~~
~~108 Third local output signal~~
~~109 Global output signal~~
~~111 Control circuit~~

~~200 Input signal~~
~~201 Capacitance~~
~~202 Inductance~~
~~203 Controllable resistance~~
~~204 Output signal~~

~~300 Wave digital filter~~
~~301 First block (serial coupler)~~
~~302 Second block (parallel coupler)~~
~~303 Third block (storage element for capacitance)~~
~~304 Fourth block (storage element for inductance)~~

~~400 Diagram~~
~~401 Abscissa~~
~~402 Ordinate~~
~~403 First curve~~
~~404 Second curve~~
~~405 Third curve~~
~~406 Fourth curve~~
~~407 Fifth curve~~
~~408 Sixth curve~~
~~409 Seventh curve~~
~~410 Eighth curve~~

~~500 Diagram~~
~~501 Abscissa~~
~~502 Ordinate~~
~~503 First curve~~
~~504 Second curve~~
~~505 Third curve~~
~~506 Fourth curve~~
~~507 Fifth curve~~

~~600 Circuit arrangement~~
~~601 First resonator circuit~~
~~602 Second resonator circuit~~
~~650 Wave digital filter~~
~~651 First component~~
~~652 Second component~~

~~700 Circuit arrangement~~
~~701 First resonator circuit~~
~~702 Second resonator circuit~~
~~703 Operational amplifier~~
~~703a Non-inverting input~~
~~703b Inverting input~~
~~703c Output~~
~~750 Wave digital filter~~
~~751 First component~~
~~752 Second component~~

~~800 Speech recognition system~~
~~801 Feature extraction system~~
~~802 Speech recognition device~~
~~803 Input signal~~
~~804 Hearing auditory channel model components~~
~~805 Signal~~
~~806 Middle ear model component~~
~~807 Signal~~
~~808 Inner ear model component~~
~~809 Signal~~
~~810 Sensor cell model component~~
~~811 Signal~~
~~812 Synaptic model component~~
~~813 Signal~~

~~901 Inductance~~
~~902 Resistance~~
~~903 Filter bank~~
~~904a Filter stage inductance~~
~~904b Filter stage resistance~~
~~904c Filter stage capacitance~~
~~905 Filter stage~~
~~905a Filter stage inductance~~
~~905b Filter stage resistance~~
~~905c Filter stage capacitance~~
~~906a Filter stage inductance~~
~~906b Filter stage resistance~~

~~906c Filter stage capacitance~~
~~907 Terminating impedance~~
~~x_{BM} Basilar membrane signal~~

~~1000 Diagram~~
~~1001 Abscissa~~
~~1002 Ordinate~~
~~1003 Excitation curve~~
~~1004 Excitation curve~~
~~1005 Excitation curve~~
~~1006 Excitation curve~~
~~1007 Excitation curve~~
~~1008 Excitation curve~~
~~1009 Excitation curve~~
~~1010 Point of greatest sensitivity~~
~~1011 Excitation threshold~~

~~1100 Filter output processing circuit~~
~~1101 High-pass filter~~
~~1102 Rectifier circuit~~
~~1103 Low-pass filter~~
~~1104 Activation circuit~~
~~1105 Vesicle circuit~~
~~1106 Neurotransmitter circuit~~
~~1107 High-pass filter capacitance~~
~~1108 High-pass filter resistance~~
~~1109 Low-pass filter resistance~~
~~1110 Low-pass filter capacitance~~

~~1200 Diagram~~

~~1300 Diagram~~
~~1301 Word error rate curve~~
~~1302 Word error rate curve~~
~~1303 Word error rate curve~~
~~1304 Abscissa~~
~~1305 Ordinate~~